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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Group Art Unit: 2644
Examiner: Walter F. Briney, III
Confirmation No.: 8830

In Re PATENT APPLICATION Of:

Appellant:	Eiichi Nishimura	SUPPLEMENTAL BRIEF ON APPEAL
Serial No.:	09/963,499	
Filed:	September 27, 2001	
For:	ECHO CANCELER WITH AUTOMATIC GAIN CONTROL OF ECHO CANCELLATION SIGNAL	
Attny Ref.:	MAE 266	February 21, 2006

Mail Stop Appeal Brief
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir:

A Notice of Appeal was filed on October 11, 2005, and a Brief was filed on November 29, 2005. This Supplemental Brief is in reply to the Notification of Non-Compliant Brief which was mailed on February 10, 2006. No fee is due. However, please charge our Deposit Account No. 18-0002 if any fees are needed to enter this paper, and please advise us accordingly. It is noted that no petition is required because of the authorization to charge, but please consider this paper a petition for extension of time if needed. This paper also appeals from the Final Official Action mailed on June 14, 2005.

FEE ENCLOSED:$ 0
Please charge any further fee to our Deposit Account No. 18-0002

REAL PARTY IN INTEREST

The real party in interest is Oki Electric Industry Company, Ltd., 7-12 Toranomon 1-chome, Minato-ku, Tokyo, Japan

RELATED APPEALS AND INTERFERENCES

To the best of the undersigned attorney's belief and knowledge, no other appeals or interferences are or have been filed which may be related to, directly affect or be directly affected by or have a bearing on the Board's decision in the pending appeal.

STATUS OF THE CLAIMS

Claims 9-15 are pending, finally rejected, and appealed. Claims 9, 13, and 14 are independent. Claims 1-8 were canceled.

STATUS OF AMENDMENTS

The final Office Action was mailed on June 14, 2005. The Appellant submitted a response on September 20 but no amendments were proposed. The claims stand as they were amended on January 21, 2005.

SUMMARY OF CLAIMED SUBJECT MATTER

The invention concerns eliminating echoes that distract listeners or garble speech, and howling or whistling caused by feedback.

With older telephone handsets, there was not much problem because the user's mouth was close to the microphone and the loudspeaker was close to the user's ear. The sound from the loudspeaker did not have to be loud, and the microphone was not sensitive, so sound from the other end of the line was not picked up at the microphone six inches from the loudspeaker.

Today the microphone and the loudspeaker may be on a table or a dashboard, instead of next to the mouth and ear, or the microphone may be on a wire dangling from the ear. This causes the voice of the other party, coming from the loudspeaker, to be picked up at the microphone with an amplitude great enough to create a "feedback loop" with howling or whistling noises, just like those of a public address amplifier turned up too loud. Phones today require echo canceling circuits that work by removing the "echo" of the other party from the microphone signal (specification page 1, 3rd paragraph).

Because of this, modern phones have an "AGC" (Automatic Gain Control) that adjusts the microphone output to stay at a constant amplitude (paragraph spanning pages

1-2). ("Gain" means "amount of amplification.") This may be needed especially if the microphone is on a cord dangling from the ear, so that it constantly moves toward and away from the mouth.

Fig. 9. Fig. 9 shows the Appellant's prior art for dealing with this situation. The AGC mentioned above is performed by the AGC unit 22 inside an automatic level adjuster 2, which sends the volume-adjusted microphone signal to an echo canceler 1. Generator 21 above the AGC unit 22 turns the AGC unit on and off, via signal level data LD, depending on the amplitude of the microphone signal TXi (for transmit signal, input). Under quiet conditions, the gain is left unchanged, effectively removing the AGC unit 22 (last lines on page 3). When above a certain level (e.g., when the user is talking) the amplitude is adjusted to keep a consistent level.

The canceler 1 takes the signal from the other party (RXi for receive signal, input) and the signal from the microphone level adjuster (TXm, amplified transmit signal) and from these two signals creates an echo cancellation signal EC (in EC signal generator 11).

The signal TXm has two components: a non-echo component from sound in the general environment (for example, the user's voice) and an echo component originating in the loudspeaker 3, which arrives at the microphone 4 through the air as echo E. Ideally, the echo component should be eliminated from the signal TXo (for transmit signal, out), to cut the feedback loop. To do eliminate it, the cancellation signal EC should be equal in amplitude to that portion of the signal TXm which comes from the loudspeaker 3, and also opposite in phase, so that it will cancel that portion of the signal TXm which comes from the loudspeaker 3 when they are electronically added together (this happens in the circled plus sign under the EC signal generator 11). The phase opposition is shown by the negative sign and the positive sign near the circled plus sign.

Coefficients. To generate the cancellation signal EC is not straightforward. It typically accomplished by adjusting parameters called "predictive filtering coefficients" (specification spanning pages 2-3) and some time is required for these parameters to "converge" to the proper values for effective cancellation (page 3, lines 7-12). The cancellation signal EC must be adjusted not only in amplitude and phase, but also in timing (because of the time it takes the sound to travel from the loudspeaker 3 to the

microphone 4, which is only a fraction of a second but is long on the electronic time scale). Because the predictive filtering coefficients are specific to each set of conditions, they must be re-calculated and must re-converge every time that the acoustic conditions change (page 3, lines 13-17).

In the prior art, the predictive filtering coefficients are adjusted only when the microphone signal TXi is greater than a certain minimum amplitude (page 3, 8th through 4th lines from the bottom), that is, *when it is subject to AGC action.* The AGC changes the gain every time that the amplitude of the microphone signal changes. For example, when the user is speaking the amplitude may go up and down rapidly. As the AGC corrects for this, the amplitude of the echo component in the signal TXm also goes up and down, even if its amplitude in front of the loudspeaker 3 is constant.

Gist. The result of the gain variation is that the amplitude of the echo component of the microphone signal constantly goes up and down during the very time that the EC signal generator 11 is generating the predictive filtering coefficients. This throws off the coefficients even as they are being calculated; there is no time for the coefficients to converge before the conditions change and a new convergence must be started. Since the coefficients cannot converge, echo cancellation is poor and the echoes degrade the sound quality.

The Appellant's specification (bottom of page 3) states this problem: "A problem arising in [prior-art] FIG. 9is that the varying gain [of the microphone signal] interferes with the accurate operation of the echo canceler 1." Echo reappears if, for example, the microphone has moved when the user again begins to speak (page 3, first full paragraph).

The Appellant addresses this problem by converging the coefficients only when the AGC is ***not*** active, i.e., when the user is ***not*** talking (second paragraph on page 5).

This solution is simple, but it is not disclosed or suggested in the applied prior art. The applied art both recognizes the problem of how long coefficient convergence takes, and attempts to solve this problem with different solutions to the coefficient convergence problem. This is discussed further below, starting on page 9.

The Appellant also has other novel and useful features.

Two AGCs. The paragraph starting on the last line of page 4, and the second full paragraph on page 5, discuss another of the Appellant's innovations: the use of two AGC

units. Both of the AGC units operate based on a common signal level, and preferably have the same gain. This is explained below.

Signal Level. Another feature is that the signal level is not adjusted while the phone is receiving. This is explained below.

The subject matter of claim 14 is explained first:

Claim 14. As exemplified by the Appellant's Fig. 1, claim 14 recites:

An echo canceler [31] *receiving a transmit signal* [Txi, page 6, line 4; from microphone 4] *and a receive signal* [RXi, page 6, line 5; from the other party], *the transmit signal including an echo* [E] *of the receive signal, comprising:*

an echo cancellation signal generator [14, page 6, line 10] *generating an echo cancellation signal* [EC] *from the receive signal by use of filter coefficients, and updating the filter coefficients when the transmit signal* [TXi] *is less than a first minimum input level and the receive signal* [RXi] *exceeds a second minimum input level;*

Condition for Updating Coefficients. The preceding clause states that the filter coefficients are updated when the signal from the microphone is less than a fixed minimum (e.g., when the user is not talking) and the signal from the other party is greater than a second fixed minimum (e.g., when the other party is talking). Figs. 3A-3D illustrate these minima. TXi(min) and RXi(min) are the minimum input levels of the transmit and receive signals, respectively. (Each figure is a "power spectrum" graph of the signal, with signal power in dBm (milliwatt decibels) on the vertical axis and frequency in Hz (cycles per second) on the horizontal axis, showing that the signal has power spread evenly over a wide range of frequencies; see page 7, third full paragraph.)

Fig. 3A represents when the microphone signal is above the minimum (condition "a"); Fig. 3B represents when the received signal is above the minimum (condition "b"); Fig. 3C represents when the microphone signal is below the minimum (condition "c"); and Fig. 3d represents when the received signal is below the minimum (condition "d"). Only under both conditions (c) and (b) are the coefficients updated: that is, when the microphone signal is low and the other party's signal is high, for example, when the other party is speaking and the user is listening. This is shown in the table of Fig. 4, and explained in the last paragraph on page 8 and the first two full paragraphs on page 9.

Signal Level Data. The graphs of Fig. 3 are also related to the signal level LD mentioned above; claim 14 continues,

a signal level data generator [24] *generating signal level data* [LD] *for the transmit signal and updating the signal level data* [LD] *when the transmit signal exceeds the first minimum input level* [TXi(min)] *and the receive signal is less than the second minimum input level* [RXi(min)] , *the signal level data being left unchanged when the receive signal exceeds the second minimum level;*

Fig. 1 shows that the signal level data generator 24 receives both the receive signal RXi from the other party and the transmit signal TXi from the microphone. How it generates the signal level data LD is explained in more detail in Figs. 2A-2E and the specification at page 7, line 11, to page 13, line 12. While Figs. 3A-3D each shows just one dBm threshold, each of Figs. 2A-2E shows two: TXi(min), which also appears in Figs. 3A and 3C; and TXi(ref), which is a target or reference level (page 7, third full paragraph). The signal level data generator 24 lets the signal stay as it is, if it is less than TXi(min); but if it exceed TXi(min) then it adjusts the amplitude to be equal to TXi(ref). Thus, the signal level data LD is either low, or at a constant amplitude (last paragraph on page 7), so that it is "left unchanged when the receive signal exceeds the second minimum level" according to claim 14.

Fig. 4 summarizes. It shows that LD is changed only under conditions (a) and (d); for example, when the other party is listening and the user is speaking; conversely, the condition for updating the filter coefficients is exemplified by the other party speaking and the user listening.

Therefore, <u>the filter coefficients are not updated while the signal level data is changing</u>. See page 8, line 13 to page 9, line 15. This is discussed further below.

AGC Units. Another feature is the use of two AGC units with specific couplings. Claim 14 continues,

a first automatic gain control unit [12] *coupled to the echo cancellation signal generator* [14], *amplifying the echo cancellation signal* [EC] *with a first gain responsive to the signal level data* [LD], *and updating the first gain when the signal level data generator* [24] *updates the signal level data, thereby generating an amplified echo cancellation signal* [ECm];

a second automatic gain control unit [23] *coupled to the signal level data generator, amplifying the transmit signal* [TXi] *with a second gain responsive to the signal level data* [LD], *and updating the second gain when the signal level data generator updates the signal level data* [24], *thereby generating an amplified transmit signal* [TXm]; *and*

an arithmetic unit [13] *coupled to the first automatic gain control unit* [12] *and the second automatic gain control unit* [23], *subtracting the amplified echo cancellation signal* [EC] *from the amplified transmit signal* [TXm], *thereby generating a transmit output signal* [TXo] *for output from the echo canceler* [31].

The only two inputs to the arithmetic unit 13 are the two AGC units 12 and 23 (Fig. 1). These units, both being "responsive to the signal level data " LD, operate in parallel; they adjust their respective first and second gains up or down together. Since (as discussed above) the signal LD is either low or held near TXi(ref), the two AGC units 12 and 23 are, basically, either both on or both off. Preferably, they both have the same gain (page 6, last full paragraph), which is a feature that is recited in dependent claim 15.

The honorable Board is invited to note that <u>the filter coefficients and the gains of the AGC units are never adjusted at the same time</u>. The specification at the bottom of page 11 points out that, because of this, gain variations do not interfere with the convergence of filter coefficients, and convergence delays are avoided.

Another effect is that the gain is not adjusted while a strong echo is present, so variations in the strong echo do not affect the AGC and the outgoing voice signal is held at the desired, constant level. This also advances the art. The Board is referred to the top of page 12 in the specification.

Figs. 5-8. Figs. 5-8 illustrate the four different states of Fig. 4 that are discussed above, exemplified by combinations of dashed and solid arrows pointing at the microphone 4. The specification between the middle of page 9 and the end of page 11 explains each drawing in detail. The advantages of the invention are listed in the three paragraphs starting at the bottom of page 11.

Independent Claim 13. Claim 13 is generally similar to claim 14 but recites fewer features. In fact, claim 14 could have been drafted as a claim depending from

claim 13, because each feature of claim 13 is also recited in claim 14. Appendix B shows claim 14 with underlining to indicate those features not recited in claim 13.

Claim 13 reads:

An echo canceler [31] *receiving a transmit signal* [TXi] *and a receive signal* [RXi], *the transmit signal including an echo* [E] *of the receive signal, comprising:*

an echo cancellation signal generator [14] *updating filter coefficients when the transmit signal is less than a first minimum input level* [TXi(min)] *and the receive signal exceeds a second minimum input level* [RXi(min)];

As is discussed above, this feature is illustrated in Figs. 3B, 3C, and 4.

a signal level data generator [24] *updating signal level data* [LD] *when the transmit signal exceeds the first minimum input level* [TXi(min)] *and the receive signal is less than the second minimum input level* [RXi(min)];

As is discussed above, this feature is illustrated in Figs. 3A, 3D, and 4.

a first automatic gain control unit [12] *updating a first gain when the signal level data generator updates the signal level data; and*

a second automatic gain control unit [23] *updating a second gain when the signal level data generator updates the signal level data.*

Claim 9. Independent claim 9 is a method claim. It recites

A method of canceling an echo [E] *of a receive signal* [RXo or RXi] *in a transmit signal* [TXo] *while controlling a signal level of the transmit signal, comprising the steps of:*

(a) detecting activity of the transmit signal and the receive signal;

(b) generating signal level data [LD] *for the transmit signal;*

(c) updating the signal level data [LD] *when the transmit signal is active and the receive signal is inactive* [second line in the first two columns of Fig. 4], *the signal level data being left unchanged when the receive signal is active* [first and third lines in the first two columns of Fig. 4; claim 9 uses the terms 'active' and 'inactive' instead of referring to minimum input levels];

(d) generating an echo cancellation signal [EC] *from the receive signal* [in EC signal generator 14, Fig. 1];

(e) amplifying the echo cancellation signal [in 1st AGC unit 12] *according to the signal level data* [LD is an input to the AGC unit 12] *, thereby generating an amplified echo cancellation signal* [ECm]*;*

(f) amplifying the transmit signal [TXo] *according to the signal level data* [], *thereby generating an amplified transmit signal* [TXm]*; and*

(g) subtracting the amplified echo cancellation signal from the amplified transmit signal, thereby generating a transmit output signal [TXo].

The operation is explained above in the discussion of claim 14.

Claim 10. This claim, depending from claim 9, recites the use of the coefficients, discussed above, for generating the EC signal, and also updating the coefficients when the transmit signal is inactive and the receive signal is active.

Claim 11. This claim, also depending from claim 9, recites that in the step of detecting activity of the transmit signal and the receive signal includes comparing the transmit signal with a first minimum input level and comparing the receive signal with a second minimum input level. This subject matter is shown in Figs. 3A-3D, as discussed above.

Claim 12. This claim is analogous to claim 15. It recites identical gain factors in the amplifying steps.

GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL

(1) Claims 9-12 were rejected under 35 USC §103 over Horna, US Patent 4,600,815, and Lane, US Patent 6,381,224.

(2) Claims 13-15 were rejected over under 35 USC §103 over Horna and Lane in view of Li, US Patent 6,580,795.

ARGUMENT: OBVIOUSNESS OF CLAIMS 13-15 OVER HORNA, LANE, AND LI

The Prior Art Recognizes the Convergence Problem But Presents Different Solution.

Horna identifies speech and echo components 110, 120 of the microphone signal (col. 4, line 13), and Horna recognizes the convergence problem at col. 1, line 64 to col. 2, line 17, stating that "the dynamic range of the speech/echo signal [must be] limited to the range of operations of the AFIR [adaptive finite impulse response] filter" (at col. 2, line 16). Horna claims that, with its invention, "the adaptive parameters of the AFIR filter will not be surpassed" (col. 3, line 31). Attenuator 32 compresses the signal to stay within the parameters of the filter (col. 4, line 15). Fig. 3 shows that all of the signal paths include feedback loops with attenuators (variable resistors) controlled by AGC units.

The Appellant claims two AGC units both receiving the same control signal LD, with one receiving a signal (EC) from the EC signal generator 14. In contrast, Horna places one AGC, 301, upstream of the adaptive filter 14, rather than downstream as with the Appellant's EC signal generator 14 and AGC 12. The other AGC, 302, controls two attenuators 32 and 33, which are in circuit positions similar to the AGCs 12 and 23 of the Appellant. Horna's AGC 302 corresponds in position to the claimed signal level data generator.

Horna states: "It is important to note that [because attenuators 32, 33 have the same gain] *the AFIR filter does not need to change value of its coefficients* in order to cancel the echo" (col. 4, lines 26-36, emphasis added). This is also the main feature of Horna's claim 1 (Horna col. 5, lines 16-24). Horna presents equal attenuation at 32 and 33 as the complete solution to the convergence problem, with the possible addition of attenuator 34 that has a gain that is "inversely proportional" to the gains at 32 and 33 (col. 4, lines 44-52).

Horna was cited during the prosecution of the other applied reference, Lane. Lane also discusses the convergence problem of its AFIR filter (which it calls the "initialization" problem) at col. 1, lines 51-67. At col. 6, line 1, Lane states, "What is then needed is [to solve the] initialization problems described above." Lane also states

(Abstract, line 5) that its invention is "to avoid the need to adapt the coefficients of an echo canceler ... as the volume level changes."

Like Horna with its attenuator 34, Lane discloses an inverted gain. However, Lane's invention is principally based on applying inverted gains to the transmit and receive signals in AGCs 53 and 63 in Fig. 3: to wit, gains G and G^{-1}; no gain; and gains f(G) and $f^{-1}(G)$.[1]

Lane explains that by applying both the gain and its inverse, they cancel out, whereby the "transfer function ... becomes $H_A(S) * G * G^{-1}$, which is equal to $H_A(S)$" (col. 5, line 25). Clearly, from the preceding statement, $G*G^{-1}$ must equal one. (A "transfer function" is a mathematical description of how a system affects signals passing through it; here, Lane is saying the signals are not affected.) The idea is similar to that of Horna, which teaches that because the filter cannot cope with change, one should remove all change from the system.

Lane's Three Regions. Lane, starting at col. 3, line 61, distinguishes three signal modes: LISTEN (the user is listening, signal energy is in the microphone signal); TALK (the user is talking, signal energy is in the speakerphone signal); and DOUBLE-TALK (mixed).

Fig. 2 shows the basis of this classification, a graph with axes E_T and E_R, quantities representing the energy in the transmit and receive signals respectively (col. 3, lines 61-66). The honorable Board will appreciate that the lines T_T and T_R in Fig. 2, which separate the regions, each corresponds to a *ratio* E_T/E_R of the transmit and receive levels. From analytic geometry, any straight line (such as T_T or T_R) is described by the formula $y = mx$, where m is the slope of the line between the y and x axes, and shows a constant ratio of y to x, because $m = y/x$. Therefore, the LISTEN, TALK, and DOUBLE-TALK regions of Fig. 2 represent specific ranges of a *ratio* of the transmit and receive signals, and not any magnitude of either one. Within the TALK region, for example, there is a high ratio of transmit to receive signal energies.

Lane modifies the "prior art" Fig. 2 in its Fig. 4 (showing its invention), and also shows the modified scheme in Fig. 5. The subject matter that Lane teaches in Fig. 4 does

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[1] Examples of these are gains 2 and ½; both gains =1; and gains (G+1)/2 (see col. 5, line 40) and its inverse, 2/(G+1), respectively.

not even use ratios of signal energies, but instead uses ratios of probabilities of signal levels (col. 6, line 7).

The Board is invited to note that the horizontal axis in Fig. 5 shows a ratio. (The vertical axis is a probability of the ratio on the horizontal axis, see col. 6, line 30).

Lane's AGCs. Both of Lane's AGC units 53 and 63 operate with a gain that depends on the mode. In TALK mode, "conventional AGC gain G" is applied by AGC 53 while the other AGC 63 applies the inverse gain G^{-1} so that the net gain in the system is zero (amplification unity), as discussed above (col. 5, lines 21-32). In LISTEN mode there is no AGC action, with both channels having unity gain (col. 5, lines 33-35). In DOUBLE-TALK mode the gain at AGC 53 is defined by a function, defined at col. 5, line 40 (equation 2), and the inverse function is applied by the other AGC (col. 5, lines 35-44).

There is no connection between the modes and the AFIR filters. There is no AGC unit for the echo cancellation signal out of filters 28 and 34 (i.e., no equivalent to the Appellant's AGC 12).

Li. Li also uses a ratio, rather than comparison to a pair of thresholds. Li is based on Lane[2] and only presents a variation on what Lane presents.

Where Lane discloses its talk, double-talk, and listen regions, Li discloses a fourth region, "silence" (compare Figs. 2 and 4 of Li). Li states that the silence region "represents the presence of low signal energy on both the transmit signal and the receive signal" (col 6, line 13). However, the Appellant believes that Li's thresholds are not magnitude thresholds, but instead are ratio thresholds. Li states: "Silence is detected when the *ratio* of b(n)/w(n) falls below a threshold value labeled 'c(n)'." (col. 6, line 40; emphasis added). The quantities b(n) and w(n) are terms in a temporal series; see col. 6, lines 28-39.

It is noted that even if the silence zone were determined by magnitudes rather than ratios (not admitted), this would correspond to the case (c) + (d) in the Appellant's Fig. 4, and not to what is claimed by the Appellant.

2 Figs. 1-2 of Lane, labeled "prior art," are copies of Lane's Figs. 1-2, and about ten paragraphs of Lane's text is copied in Li. Lane is cross-referenced by Li at col. 1, line 15, and John Lane, of the Lane patent, is the co-inventor of the Li patent. Both John Lane and Zhao Li work for Motorola and live in Austin.

Li calls the transmit signal magnitude "x" and the receive value magnitude "y" (col. 7, lines 31-34). The ratio is determined on the basis of a comparison of the values of the transmit signal "x" and the receive value "y" to one another (steps 268 and 274 in Fig. 7). The Li "listen" state is detected by setting listen flag to "1" (step 262 in Fig. 7) or clearing the flag to "0." Li also uses a pair of counting variables "k" and "n".

Features of Claim 13 Are Not Disclosed in Any of the References. Claim 13 recites the following features not found in Horna, Lane, or Li:

(1) *...an echo cancellation signal generator updating filter coefficients when the transmit signal is less than a first minimum input level and the receive signal exceeds a second minimum input level ...*

The Examiner admits (page 5, lines 4-7) that this is not disclosed by Horna (final Office action at page 2, last two lines). Indeed, Horna discloses that the coefficients of its AFIR filter are *never* updated—they don't need to be, it says, because its attenuators 32 and 33 have the same gain (col. 4, lines 27-36, quoted above).

Only a maximum input receive level is disclosed, contrary to the claim. Horna states (col. 3, line 66) that device 301 will "attenuate signals of abnormally high amplitudes," i.e., the filter sees only signals less than a maximum, not those greater than a minimum The Examiner has not cited to Horna for any minimum in the receive signal. As to the transmit signal, a portion that reaches the AFIR filter 14 is the "ERROR" through attenuator 34, which is not the actual transmit signal because it comes through the inverse attenuator 34. Whatever the actual transmit signal is, the filter 14 does not see it. Therefore the filter 14 cannot possibly respond to any minimum signal, and, in any case, there is not even a disclosure of any minimum in the inverse-gain signal that the AFIR does see.

Lane. Neither are these features is disclosed by Lane. Just like Horna, Lane claims that the coefficients do not need to be updated, this time due to the inverted gains of AGCs 53 and 63 instead of due to attenuators 32, 33. As in Horna, there is no disclosure of any minimum input *levels*, there is only disclosure of *ratios* E_T/E_R between the receive and transmit signals, as is discussed above, and a minimum of a ratio does not imply any particular level or amplitude of either of its components (i.e., the ratio 8/32 is the same as the ratio 1/4, but 8 does not equal 1).

Lane's ratios do not even *contain* any minima that could possibly anticipate the Appellant's claimed clause above. In Lane's Fig. 4, all of the regions converge to the origin and no actual magnitude of either signal energy is disclosed.

In spite of the fundamental mathematical difference between what is claimed and what is disclosed, the Examiner maintains that they are the same (bottom of page 8, final Action). The Examiner might as well assert that saying "This dog is twice as big as that one" is the same as saying that "This dog weighs more than 30 pounds and that one weighs less than 15 pounds." Under the Examiner's logic, the person of ordinary skill would buy a tiny lapdog because his neighbor advised him that he should have two dogs, one weighing twice as much as the other.

Furthermore, as noted above, Lane teaches using the *probability* of a ratio of signal energies (Fig. 4) rather than actual energy levels ("prior art" Fig. 2). If a person of ordinary skill in the art had combined the references (not admitted obvious), he or she would have used the teachings of Fig. 4 and col. 6, line 7 *ff*. Thus, the rejection is perhaps more analogous to the person of ordinary skill buying a tiny lapdog because his neighbor advised him that he is likely to obtain two dogs, one weighing twice as much as the other.

Of Lane's regions/modes, that which is closest to the Appellant's clause above (which corresponds to a low transmit signal and a high receive signal as exemplified in line 3 of Fig. 4), would be the LISTEN region. (There is no actual correspondence because no actual minima are disclosed by Lane). In the LISTEN mode, there is unity gain (Fig. 3 and col. 5, line 32) and therefore there is no change in the receive and transmit signal levels. In the LISTEN mode, Lane's AGCs 53 and 63 could be physically removed from the circuit and nothing would change. The honorable Board is invited to consider: with the AGCs replaced by straight-through lines, what in Lane's Fig. 3 would suggest detecting a minimum of either the signal? What would suggest using that minimum to control the AFIR filters? What would suggest not one, but two minima each affecting just one filter? The Appellant sees no such suggestions.

Li. As is noted above, Li is just a variation on Lane and, like Lane, presents nothing that would anticipate the clause above.

The honorable Board is invited to consider Li's "silence region" in Fig. 4. It shows a region in which *the transmit signal is less than a first minimum input level and the receive signal* *<u>is less than</u>* *a second minimum input level*, not a region in which *the transmit signal is less than a first minimum input level and the receive signal* *<u>exceeds</u>* *a second minimum input level*, which is what is claimed. In Li's Fig. 4, the claimed feature would be shown by a rectangular area lying on the horizontal axis, located to the right of the "silence region," and extending off to infinity along the axis.

Li teaches adjusting gains when not in the silence region, but when in the silence region it does nothing. "If gain factor controller 120 detects silence, it bypasses the [other three states'] determination, saving many valuable processing steps" (col. 5, line 62; see also col. 7, line 13). Fig. 8 also shows that coefficients are not updated in the silence state.

(2) *... a signal level data generator updating signal level data when the transmit signal exceeds the first minimum input level and the receive signal is less than the second minimum input level ...*

The Examiner admits (bottom of page 2) that Horna does not disclose updating signal level data (LD) when (in the language of claim 9) the transmit signal is active and the receive signal is inactive (i.e., the conditions of Appellant's Figs. 3A and 3D, as tabulated in line 2 of Fig. 4). The Examiner relies on Lane for disclosing this feature, but the Appellant respectfully disagrees.

First, Lane has nothing analogous to the claimed signal level generator. The Appellant does not see in Lane any explanation whatsoever of how or by what the signal level ratios might be measured.

It appears likely that the two AGCs 53 and 63 could perform these measurement, simply because the other parts—conventional AFIR filters 28/34, A/D converters 51/52, and summing junctions 25/32—seem less likely, and it is the AGCs that respond to the measured ratios. However, the Appellant claims two AGCs separately (see below) and it would be improper to apply only one feature of Lane to anticipate two distinct features of the instant claims.

(3) *... a first automatic gain control unit updating a first gain when the signal level data generator updates the signal level data; and*

a second automatic gain control unit updating a second gain when the signal level data generator updates the signal level data.

Assuming that Lane actually did disclose a signal level generator (not admitted), in order to anticipate it would have to operate "when the transmit signal exceeds the first minimum input level and the receive signal is less than the second minimum input level," a condition that corresponds to line 2 of the Appellant's Fig. 4. As is noted above, Lane does not actually disclose any levels, only ratios, and it teaches using probabilities of ratios of energies: however, in Lane's TALK region the transmit signal energy E_T is greater than receive signal energy E_R. In this mode the gains are the inverses, G and G^{-1} (see Fig. 3 of Lane).

This would be directly contrary to the Appellant's claim 15.

As to claims 13 and 14, the Applicant points out that these claims recite just one signal level data generator, but two AGCs. However, as was pointed out above, the only feasible location for a signal level data generator in Lane is within the AGCs, and there is no connection between them, so there would need to be two signal level data generators in Lane. While a second signal level data generator is not ruled out (due to "comprise" in the claim), the claim language requires that both AGCs can respond to "the" one signal level data generator; the claim covers the case of one signal level data generator.

Claims 14 and 15. Claim 14, which contains all the features of claim 13, is patentable for the reasons above. Claim 14 recites an additional feature, *the signal level data being left unchanged when the receive signal exceeds the second minimum level,* which does not appear in Lane because Lane discloses only a ratio involving the receive signal, and probabilities of a ratio involving the receive signal. Neither do the other references disclose this.

Claim 15 depends from claim 14 and its features are directly contrary to the inverses G and G^{-1} of Lane.

ARGUMENT: OBVIOUSNESS OF CLAIMS 9-12 OVER HORNA AND LANE

The arguments for apparatus claims 13-15 above also apply to method claims 9-12. For example, claim 12 corresponds to claim 15 and is patentable for a similar reason. In particular, claim 9 recites ... *(b) generating signal level data for the transmit signal;*

(c) updating the signal level data when the transmit signal is active and the receive signal is inactive, the signal level data being left unchanged when the receive signal is active ... which are analogous to the features of claims 13 and 14, as the honorable Board will appreciate.

In summary, a major difference between the Appellant's claimed subject matter and the applied art is, *inter alia*, that the Appellant generates an echo cancellation signal according to the levels of the receive and transmit signals, so that the coefficients needed for echo cancellation are only updated when the signal levels are appropriate. The Examiner asserts that Lane discloses this feature, but Lane only discloses changing the gains of the AGCs based on probabilities of ratios of signal levels, not the levels themselves, contrary to the claims. Also, the gains do not affect the filters of Lane, which are downstream of the AGCs. Neither Horna nor Li disclose this feature either.

Respectfully submitted,

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APPENDIX A—CLEAN CLAIMS

1-8. (canceled)

9. (original): A method of canceling an echo of a receive signal in a transmit signal while controlling a signal level of the transmit signal, comprising the steps of:

(a) detecting activity of the transmit signal and the receive signal;

(b) generating signal level data for the transmit signal;

(c) updating the signal level data when the transmit signal is active and the receive signal is inactive, the signal level data being left unchanged when the receive signal is active;

(d) generating an echo cancellation signal from the receive signal;

(e) amplifying the echo cancellation signal according to the signal level data, thereby generating an amplified echo cancellation signal;

(f) amplifying the transmit signal according to the signal level data, thereby generating an amplified transmit signal; and

(g) subtracting the amplified echo cancellation signal from the amplified transmit signal, thereby generating a transmit output signal.

10. (original): The method of claim 9, wherein said step (d) is carried out by use of coefficients, further comprising the step of:

updating the coefficients when the transmit signal is inactive and the receive signal is active.

11. (original): The method of claim 9, wherein said step (a) further comprises the steps of:

comparing the transmit signal with a first minimum input level; and

comparing the receive signal with a second minimum input level.

12. (original): The method of claim 9, wherein said step (f) and said step (g) employ identical gain factors.

13. (previously presented): An echo canceler receiving a transmit signal and a receive signal, the transmit signal including an echo of the receive signal, comprising:

an echo cancellation signal generator updating filter coefficients when the transmit signal is less than a first minimum input level and the receive signal exceeds a second minimum input level;

a signal level data generator updating signal level data when the transmit signal exceeds the first minimum input level and the receive signal is less than the second minimum input level;

a first automatic gain control unit updating a first gain when the signal level data generator updates the signal level data; and

a second automatic gain control unit updating a second gain when the signal level data generator updates the signal level data.

14 (previously presented): An echo canceler receiving a transmit signal and a receive signal, the transmit signal including an echo of the receive signal, comprising:

an echo cancellation signal generator generating an echo cancellation signal from the receive signal by use of filter coefficients, and updating the filter coefficients when the transmit signal is less than a first minimum input level and the receive signal exceeds a second minimum input level;

a signal level data generator generating signal level data for the transmit signal and updating the signal level data when the transmit signal exceeds the first minimum input level and the receive signal is less than the second minimum input level, the signal level data being left unchanged when the receive signal exceeds the second minimum level;

a first automatic gain control unit coupled to the echo cancellation signal generator, amplifying the echo cancellation signal with a first gain responsive to the signal level data, and updating the first gain when the signal level data generator updates the signal level data, thereby generating an amplified echo cancellation signal;

a second automatic gain control unit coupled to the signal level data generator, amplifying the transmit signal with a second gain responsive to the signal level data, and

updating the second gain when the signal level data generator updates the signal level data, thereby generating an amplified transmit signal; and

an arithmetic unit coupled to the first automatic gain control unit and the second automatic gain control unit, subtracting the amplified echo cancellation signal from the amplified transmit signal, thereby generating a transmit output signal for output from the echo canceler.

15 (previously presented): The echo canceler of claim 14, wherein the first gain is equal to the second gain.

APPENDIX B—CLAIM 14 UNDERLINED TO SHOW DIFFERENCES FROM CLAIM 13

An echo canceler receiving a transmit signal and a receive signal, the transmit signal including an echo of the receive signal, comprising:

an echo cancellation signal generator generating an echo cancellation signal from the receive signal by use of filter coefficients, and updating the filter coefficients when the transmit signal is less than a first minimum input level and the receive signal exceeds a second minimum input level;

a signal level data generator generating signal level data for the transmit signal and updating the signal level data when the transmit signal exceeds the first minimum input level and the receive signal is less than the second minimum input level, the signal level data being left unchanged when the receive signal exceeds the second minimum level;

a first automatic gain control unit coupled to the echo cancellation signal generator, amplifying the echo cancellation signal with a first gain responsive to the signal level data, and updating the first gain when the signal level data generator updates the signal level data, thereby generating an amplified echo cancellation signal;

a second automatic gain control unit coupled to the signal level data generator, amplifying the transmit signal with a second gain responsive to the signal level data, and updating the second gain when the signal level data generator updates the signal level data, thereby generating an amplified transmit signal; and

an arithmetic unit coupled to the first automatic gain control unit and the second automatic gain control unit, subtracting the amplified echo cancellation signal from the amplified transmit signal, thereby generating a transmit output signal for output from the echo canceler.